Amendments to the Claims

Claim 1 (Currently amended): A method for generating a masking threshold level for reducing code quantization in a digital audio system, the threshold comprising both simultaneous masking and temporal masking effects on an audio signal to be coded; the method comprising ~~the steps of~~:

a) providing a filter having a selected transfer function;
b) inputting simultaneous masking signals into the filter;
c) generating approximate replica temporal masking signals at the filter output;
d) adding the simultaneous masking signals and the replica temporal masking signals to form a composite masking signal; and
e) using the composite masking signal to establish the masking threshold level.

Claim 2 (Currently amended): The method recited in claim 1 further comprising ~~the steps of~~:

f) carrying out said code quantization in each of a plurality of frequency domain subbands over a broad audio bandwidth; and
g) performing steps a) through e) in each said subband.

Claim 3 (Currently amended): The method recited in claim 1 further comprising ~~the steps of~~:

f) continuously carrying out said code quantization over a plurality of sequential time frames; and

g) performing steps a) through e) over a selected number of said sequential time frames.

Claim 4 (Original): The method recited in claim 1 wherein said selected transfer function causes said temporal masking signals to decay approximately exponentially with the logarithm of time.

Claim 5 (Original): The method recited in claim 1 wherein said selected transfer function causes said temporal masking signals to decay at a rate which is approximately inversely proportional to the duration of the corresponding simultaneous masking signal.

Claim 6 (Original): The method recited in claim 1 wherein said filter is an infinite impulse response filter.

Claim 7 (Original): The method recited in claim 6 wherein said filter is an M order auto regressive and L order moving average filter.

Claim 8 (Original): The method recited in claim 7 wherein said filter is selected to have M=2 and L=2.

Claim 9 (Original): The method recited in claim 1 wherein said selected transfer function is of the form

$$H(z) \quad \frac{Az^{-1} + Bz^{-2}}{1-Cz^{-1} - Dz^{-2}}$$

where A .25, B 0.06. C 0.39 and D 0.295.

Claim 10 (Original): The method recited in claim 2 wherein step g) is carried out in fewer than the total number of subbands in said plurality of subbands.

Claim 11 (Currently amended): A method for reducing quantization coding bits in a digital audio system by employing a masking threshold level that includes the effects of both simultaneous masking and temporal masking over a plurality of time frames; the method comprising ~~the steps of~~:

a) providing a filter which has a selected transfer function for simulating temporal masking decay that is exponential with the logarithm of time;
b) inputting simultaneous masking signals into the filter;
c) generating approximate replica temporal masking signals at the filter output;
d) adding the simultaneous masking signals and the replica temporal masking signals to form a composite masking signal; and
e) using the composite masking signal to establish the masking threshold level.

Claim 12 (Currently amended): The method recited in claim 11 further comprising ~~the steps of~~:

f) carrying out said code quantization in each of a plurality of frequency domain subbands over a broad audio bandwidth; and

g) performing steps a) through e) in each said subband.

Claim 13 (Currently amended): The method recited in claim 11 further comprising ~~the steps of~~:

f) continuously carrying out said code quantization over a plurality of sequential time frames; and

g) performing steps a) through e) over a selected

number of said sequential time frames.

Claim 14 (Original): The method recited in claim 11 wherein said selected transfer function causes said temporal masking signals to decay at a rate which is approximately inversely proportional to the duration of the corresponding simultaneous masking signal.

Claim 15 (Original): The method recited in claim 11 wherein said filter is an infinite impulse response filter.

Claim 16 (Original): The method recited in claim 15 wherein said filter is an M order auto regressive and L order moving average filter.

Claim 17 (Original): The method recited in claim 16 wherein said filter is selected to have M=2 and L=2.

Claim 18 (Original): The method recited in claim 11 wherein said selected transfer function is of the form

$$H(z) \quad \frac{Az^{-1} + Bz^{-2}}{1 - Cz^{-1} - Dz^{-2}}$$

where A .25, B 0.06. C 0.39 and D 0.295.

Claim 19 (Original): The method recited in claim 12 wherein step g) is carried out in fewer than the total number of subbands in said plurality of subbands.